Siemens AG
New PCT application
26965-2389 (P-01,0087)
1998P02433WOUS
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Re: Substitute Pages

Translation / February 23, 2001 / 1696(911) / 4060 words

METHOD AND APPARATUS FOR ADAPTING A TRANSMISSION POWER TO THE TRANSMISSION QUALITY OF A TRANSMISSION CHANNEL

The invention is directed to a method for adapting transmission power to the transmission quality of a transmission channel.

The need for digital transmission systems has exponentially risen in recent decades. Digital transmission systems are generally classified into the function units shown in Fig. 1. A message source 1 generates information that a transmitter transmits to a receiver via a transmission channel 4. The properties of the information to be transmitted are dependent on the message source. Messages to be transmitted can, for example, be an audio signal or a video signal. Analog transmission systems thereby transmit analog signals that were generated by analog message sources directly via the transmission channel upon employment of traditional analog modulation methods. Such modulation methods are, for example, amplitude modulation, frequency modulation or phase modulation. In digital transmission systems, the information to be transmitted is converted into a sequence of binary numbers. In order to be able to utilize the capacity of the channel optimally well, the message to be transmitted should be represented with as few binary numbers as necessary. To this end, a source encoder is employed that has the job of converting the messages to be transmitted into sequences of signal values and encoding them, so that the channel can transmit them. The source encoder thereby attempts to convert the messages to be transmitted into binary numerals as efficiently as possible.

The sequence of binary numbers generated by the source encoder is transmitted by the channel to the receiver. Such an actual channel can, for example, be composed of a line connection, of a coaxial cable, of a light waveguide (LWL), of a radio connection, a satellite channel or a combination of these transmission media. Such channels cannot directly transmit the sequence of binary numbers from the transmitter. To that end, the sequence of digital information must be converted into signal values that correspond to the properties of the channel. Such a device is called a digital modulator. Such a modulator is part of the channel encoder 3, which

additionally comprises a discrete channel encoder in order to provide the information to be transmitted with an error protection adapted to the channel.

It is not assumed of the transmission channel 4 that it works error-free; rather, it is assumed that a noise source 5 will modify the transmitted signals during the transmission with a specific probability.

Such disturbances can, for example, be a cross-talk of signals that are transmitted on neighboring channels. The disturbances can likewise be caused by thermal noise that is generated in the electronic circuit such as, for example, amplifiers and filters that are employed in the transmitter and in the receiver. Given line connections, disturbances can additionally be caused by switchings and can be additionally caused by meteorological influences given radio or satellite connections such as, for example, thunderstorms, hail or snow. Such influences modify the transmitted signal and cause errors in the received digital signal sequence.

In order to nonetheless assure a relatively dependable transmission, the channel encoder increases the redundancy of the (binary) sequence to be transmitted. With the assistance of this redundancy added by the transmitter, the receiver is assisted in the decoding of the information-carrying signal sequence. To this end, for example, the channel encoder combines a specific plurality of signals to form blocks and a plurality of check signals (one parity bit in the simplest case) is added. In this way, k information bits are always simultaneously encoded, whereby each k bit sequence has an unambiguous n bit sequence, what is referred to as the code word, allocated to it. The redundancy added in this way can be indicated with the ratio n/k. This likewise corresponds to the channel bandwidth that must be correspondingly increased in order to transmit the information sequence expanded by the added redundancy.

Alternatively, an enhanced dependability against channel disturbances can also be achieved, for example, by an increase in the transmission power. Since the increase in the transmission power, however, is relatively expensive, the dependability is usually achieved given available bandwidth by increasing the required channel bandwidth.

WO 97/03403 discloses the data transmission with variable data rate in a cellular radio system. Before data are transmitted via a transmission channel, they usually pass through two encoding units, namely a voice encoding and a channel encoding. The voice encoding reduces the quantity of data required for the transmission of a specific information. The channel encoding attaches further data to the data encoded in this way in order to also assure a dependable transmission given a disturbed channel. Effective voice encoding methods supply a data stream with variable data rate dependent on the information to be encoded. Such a saving of data can usually not be directly used for other data transmissions in mobile radiotelephony. The data saved by the effective voice encoding are therefore used in order to lower the transmission power of the transmitter.

In the transmission of one bit with the data rate R bit/s, the modulator always allocates a signal curve or, respectively, a signal value (referred to below only as signal value) $s_1(t)$ to the binary number 0 and allocates a signal value $s_2(t)$ to the binary number 1. This transmission of each individual bit by the channel encoder is called binary modulation. Alternatively, the modulator can simultaneously transmit k information bits upon employment of $M = 2^k$ different signal values $s_i(t)$ with $i = 1, 2, \ldots M$, whereby each of the 2^k possible k-bit sequences is allocated to a signal value.

At the receiver side of a digital transmission system, the digital demodulator processes the signal value transmitted in the channel (potentially modified) and allocates an individual number to each signal value that represents an estimate of the transmitted data symbol (for example, binary).

After reception of a signal in the receiver, the demodulator must decide which of the M possible signal values was sent. This decision is implemented in a decision unit (slicer), whereby the decision should be made with minimal error probability. This decision unit allocates a reception value (usually edited) to one of the M possible symbol values.

When, for example, a binary modulation is employed, the demodulator must decide when processing each received signal whether the transmitted bit is a matter of a 0 or of a 1. In this case, the demodulator implements a binary decision. Alternatively, the demodulator can also implement a ternary decision, whereby the

demodulator decides for "0", "1" or "no decision" dependent on the quality of the received signal.

The decision process of a demodulator can be viewed as quantization, whereby binary and ternary decisions are specific instances of a demodulation that quantizes the Q-level, whereby $Q \geq 2$ applies. In general, digital communication systems employ a high-order modulation, whereby $m = 0, 1 \ldots M-1$ represents the possible transmitted symbols.

When the transmitted information contains no redundancy, the demodulator must decide at every predetermined time interval which of the M-signal values was transmitted. When the transmitted information, in contrast, contains redundancy, then the demodulator reconstructs the original information sequence on the basis of the code employed by the channel encoder and on the basis of the redundancy of the received signals. Dependent on the demands defined by the applications, the channel encoder generates signal blocks that make it possible for the channel decoder to either only identify where the specific disturbances have occurred (error-recognizing encoding) or to even be able to automatically correct (error-correcting encoding) errors caused by disturbances (up to a specific maximum number per signal block).

One criterion for the dependability with which the messages are transmitted from the transmitter to the receiver is represented by the error rate. The error rate indicates the average probability with which a bit error occurs at the output of the decoder. The bit error rate indicates the plurality of error bits occurring at the receiver divided by the total number of received bits per time unit. The bit error rate (or symbol error rate when the error frequency of symbols is evaluated) is the most important quality criterion of a digital transmission system. In general, the error probability is dependent on the code properties, on the nature of the signal values employed for the transmission of the information via the channel, on the transmission power, on the properties of the channel, i.e. the strength of the noise, the type of noise, etc., and on the demodulation and decoding method. The significance of the bit error rate for digital transmission systems corresponds to the signal-to-noise ration (SNR) of analog transmission systems.

The error rates with which symbols occur at the output of the demodulator or, respectively, with which bits occur at the output of the decoder are dependent on the properties of the transmission medium, i.e. of the transmission channel, on the selected modulation and encoding strategy and on the average power of the transmission signal. For adaptation of a transmission data rate to a transmission channel, the transmission properties of the transmission channel are traditionally determined by communicating a bit or, respectively, symbol sequence that is known to the receiver. The error rate of the channel can be determined on the basis of a rated-actual comparison in the receiver. In this way, the quality of the current data transmission can be identified. What is disadvantageous about this method, however, is that only the measurement of a possible combination of transmission power, encoding method and modulation method can be measured. So that a separate measurement need not be implemented for every possible data rate or, respectively, transmission power, iterative methods are usually utilized for finding an optimum transmission data rate or, respectively, transmission power.

GB-A-2303769 discloses a communication equipment that is in the position of setting the transmission data rate. First, a transmission data rate is selected dependent on the measured electrical field strength. This transmission data rate is subsequently additionally varied dependent on a measured bit error rate, namely reduced given a high error rate and boosted given a low error rate. The bit error rate measurement serves as basis for a fine adjustment of the transmission data rate.

An object of the invention is to create an improved method and an improved apparatus for adapting the transmission power to the transmission channel.

This object is achieved for an apparatus with the technical teaching of patent claim 1 and is achieved for a method with the technical teaching of patent claims 6.

Inventively, a transmission power is set dependent on the measured transmission quality of the transmission channel. With the measurement of the transmission quality, particularly with a receiver-side determination of the signal-to-noise ratio based on the received signals, the transmission power can be minimized dependent on the transmission data rate employed.

In this way, the transmission sequence of modulator/transmission channel/demodulator can be measured on line (i.e., during the data transmission) independently of the selected encoding method, and the transmission power can be set such dependent on the required data transmission rate that a predetermined bit or, respectively, symbol error rate is guaranteed. The measurement of the transmission quality is the prerequisite in order to define the minimum transmission power such for a defined transmission rate that a maximally acceptable error rate is not exceeded.

The power of the transmitter can be adapted to the required transmission quality in that the transmission power is raised or, respectively, lowered dependent on a difference between a measured signal-to-noise ratio and a required signal-to-noise ratio. In this way, the transmission power, based on a measurement of the signal-to-noise ratio, can be optimally adapted, i.e. minimized, to the selected transmission method and the existing transmission channel, i.e. lowest possible transmission power given simultaneous assurance of the quality demands and adherence to the required transmission rate. The noise emissions are thus minimized and, at the same time, the transmission capacity of neighboring systems that work on the same frequency band is increased.

Advantageous developments of the invention are recited in the subclaims.

Preferred exemplary embodiments of the invention are explained below on the basis of the drawing. Shown are:

Fig. 1 the general structure of a message transmission system;

Fig. 2 the structure of an inventive transmission system for adapting the data rate and the modulation method to the transmission medium on the basis of receiver-side signal-to-noise ratio measurement.

Fig. 3 the structure of an inventive transmission system for adapting the transmission data rate, the modulation method and the transmission power to the transmission medium on the basis of receiver-side signal-to-noise ratio measurement; and

Fig. 4 a diagram for illustrating the "power control" for setting a transmission power dependent on a measure and on an employed transmission quality.

In digital information transmission, information are transmitted between a message source (transmitter) and a receiver via a transmission medium. Such an

apparatus that is located between the transmitter and the receiver is generally referred to as channel.

For the transmission, the data to be transmitted are converted into code words that are matched to the transmission properties of the message channel in order to protect the data to be transmitted against among other things, transmission errors.

signal values 21 edited in this way are subsequently supplied to a decision unit or, respectively slicer 22 that allocates a symbol 23 to every received signal value 21.

The symbol/bit converter 24 of the channel decoder 20 allocated and encoded, digital information or, respectively, an encoded bit sequence 53 to each detected symbol or, respectively, each detected symbol sequence 23 according to the selected mapping method, the digital information or, respectively, the bit stream 25 being derived therefrom with the assistance of the digital channel decoder 52 according to the selected encoding method.

The decision unit (slicer) 22 is a basic component part of every demodulator. Such a decision unit allocates the symbol or, respectively, the symbols that was most probably sent to the reception value - usually edited. Since the set of input values of the decision unit, due to disturbances or distortions of the transmission channel, usually does not correspond to the "valid" signal values of the transmitter, i.e. the signal values that are allocated to the symbols to be transmitted, the signal-to-noise ratio 28 adjacent to the decision input can be determined from the input signal 21 and the output signal 23 of the decision unit independently of the encoding and mapping algorithm employed. To this end, an inventive receiver comprises a device 27 for measuring the signal-to-noise ratio (SNR) of the information transmitted via the transmission channel 11.

In a possible embodiment of a device for measuring the signal-to-noise ratio, a signal value 60 that the input of the decision unit in the demodulator would have received if the signal curve or, respectively, signal value corresponding to the detected symbol had been transmitted unfalsified is again allocated to every detected symbol in the demodulator at the receiver side. In this way, a hypothetical input signal corresponding to the detected symbol values that contains no signal values with channel distortions or disturbances is formed. This reference signal - as long as the decision unit does not detect any incorrect symbols - thus corresponds to the original signal at the transmitter side. By subtracting this reference signal from the edited receiver signal 21, the noise signal can be acquired.

The average power of this reference signal formed in this way corresponds to the average power of the received undisturbed signal part. The average power of

the signal adjacent at the input of the decision unit corresponds to the aggregate power of received noise and signal part. The noise power is calculated therefrom with the assistance of the previously calculated, undisturbed signal part. The signal-to-noise ratio (SNR) as a criterion for the transmission quality of the transmission channel derives from the ratio of the average power of the undisturbed signal part to the average power of the noise part.

What such a method avoids is that the receiver must know a specific transmission sequence, as necessary given other, traditional methods. Moreover, the determination of the error rate ensues parallel to the evaluation of the transmitted symbols, i.e. online. A periodic introduction of a test sequence into the data stream to be transmitted is therefore no longer required for the continuous measurement of the transmission quality. In this way, a reduction of the net data rate of the transmission channel can be avoided.

In order to assure a high statistical dependability, a traditional method that employs a test sequence known to the transmitter and receiver must cover a great number of errors, usually several hundred. The traditional methods require very long measuring times in order to detect a corresponding plurality of errors for the very low bit error rates of, for example, 10^9 that are generally required. The inventive method, in contrast, is based on the interpretation of the measured signal-to-noise ratio during ongoing transmission. Since, however, only significantly shorter measuring times are required for the interpretation of the average powers when compared to the comparable interpretation of the symbol or, respectively, bit stream, the transmission quality can be determined far faster with the inventive method.

Dependent on the selected encoding and mapping method, there is always an unambiguous functional relationship between the signal-to-noise ratio 28 and a symbol error rate or, respectively, bit error rate. The signal-to-noise ratio thus qualifies the transmission properties of the channel and of the momentarily selected modulation or, respectively, demodulation method independently of the selected encoding or, respectively, mapping method. Via a measurement of the signal-to-noise ration 28 of a transmission channel 11, thus, the encoding or, respectively, the mapping method of the current modulation/demodulation method can be defined such